DEFENSE INFORMATION SYSTEMS AGENCY
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Joint Interoperability Test Command (JTE)   **21 Sep 10**

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Extension of the Special Interoperability Test Certification of the Avaya S8710 and S8720 Digital Switching Systems with Software Release Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)

References: (a) DoD Directive 4630.05, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004
          (b) CJCSI 6212.01E, "Interoperability and Supportability of Information Technology and National Security Systems," 15 December 2008
          (c) through (g), see Enclosure

1. References (a) and (b) establish the Defense Information Systems Agency (DISA), Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification.

2. The Avaya S8710 Digital Switching System with Software Release CM 4.0 (R014x.00.2.731.7: Super Patch 14419) is hereinafter referred to as the System Under Test (SUT). The SUT met all critical interoperability requirements and is certified as interoperable for joint use within the Defense Switched Network (DSN). The SUT is certified to support DSN Assured Services over Internet Protocol with any Assured Services Voice Application Local Area Network (ASVALAN) on the Unified Capabilities (UC) Approved Products List (APL). The SUT is also certified for joint use with any Voice Application Local Area Network (VALAN) on the UC APL. However, since VALANs do not support the Assured Services Requirements detailed in Reference (c), Command and Control (C2) users and Special C2 users are not authorized to be served by the SUT connected to a VALAN. The identified test discrepancies shown in the SUT Interoperability Test Summary, which remained open after Super Patch 14419 was applied and regression tested, have an overall minor operational impact. The SUT was tested and met the critical interoperability requirements for the following DSN switch types: Small End Office (SMEO), Private Branch Exchange (PBX) 1, PBX 2, and Deployable Voice Exchange (DVX). The Avaya S8720 employs the same software and hardware as the Avaya S8710 with the exception of the S8720 media server. Analysis by JITC determined that the S8720 is functionally identical to the S8710 for interoperability certification purposes, and it is also certified for joint use within the DSN as a SMEO, PBX 1, and PBX 2.

The S8700 series media servers work in conjunction with the G650 complementary media gateways which support multi-protocol environments for concurrent support of Time Division Multiplex (TDM) and Internet Protocol (IP)-based telephony. The SUT is capable of supporting

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three port networks with a maximum of five G650s on each port network. The JITC, however, conducted testing on the SUT using only two port networks, each of which had two G650s. Based on this testing and through analysis, this certification only applies to S8700 systems that are configured for utilization of two port networks with a maximum of ten G650s (five on each port network). The SUT offers an internal Automated Call Distributor (ACD), which was tested and is covered under this certification. The SUT does not offer an internal voicemail capability; however, the SUT is certified for external voicemail systems on the UC APL via the 2-wire proprietary digital interface. The SUT is certified for conferencing through an external conferencing bridge that is on the UC APL. No other configurations, features, or functions, except those cited within this report, are certified by the JITC. This certification expires upon changes that affect interoperability, but no later than three years from the date of the original memorandum (2 October 2007).

3. The extension of this certification is based upon Desktop Review (DTR) 11. The original certification is based on interoperability testing conducted by JITC and a review of the vendor's Letters of Compliance (LoC). Interoperability testing was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 29 May through 16 July 2007. Regression testing was conducted from 7 through 10 August 2007 and documented in Reference (d). Review of the LoC was completed on 13 August 2007. This DTR was requested to include the Avaya IP Softphone Release 6.0. The JITC determined there was a minor risk in approving this DTR because the Avaya IP Softphone was included in Real Time Services (RTS) testing conducted at JITC. The Avaya IP Softphone will be certified with the following client hardware/software or equivalent: Dell Latitude 510/Dell Inspiron 6000, Microsoft Windows XP with Service Pack 3, 1.73 GigaHertz Intel Pentium Processor, and 1 gigabyte of Random Access Memory. The DSAWG accreditation of this DTR was granted on 21 September 2010.

4. The SUT interoperability test summary is depicted in Table 1. The SMEO Capability Requirements (CRs) and Feature Requirements (FRs) are listed in Table 2. If a switch meets the SMEO requirements, it meets the lesser requirements of a PBX 1 and PBX 2. The comparison between SMEO and DVX requirements and interoperability status is listed in Table 3. This interoperability test status is based on the SUT's ability to meet:

   a.   DSN services for Network and Applications specified in Reference (c).

   b.   SMEO and DVX interface and signaling requirements for trunks/lines specified in Reference (e) verified through JITC testing and/or vendor submission of LoC.

   c.   SMEO and DVX CRs/FRs specified in Reference (e) verified through JITC testing and/or vendor submission of LoC.

   d.   Internet Protocol version 6 requirements specified in Reference (e), paragraph 1.7, Table 1-4, verified through vendor submission of LoC signed by the Vice President of the company.

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    e.   The overall system interoperability performance derived from test procedures listed in Reference (f).

    f.   The overall softphone requirements specified in Reference (g), paragraph 5.3.2.6.1.7.

Table 1. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
T1 CAS (DTMF, DP)	Yes	Certified	Met all CRs and FRs with the following minor exceptions: The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification.[1] The SUT T1 CAS preemption signal generation is out of tolerance.[2] The SUT recognizes E1 and T1 CAS wink start signals greater than the maximum interval as valid.[3] During a remote busy condition on a T1 CAS or E1 CAS, the SUT takes approximately 5 minutes to change the status of the timeslots from an "In-Service/Active" state to a "Far-End-Busy" state.[4]
T1 CAS (MFR1)	No	Certified	Met all CRs and FRs with the following minor exceptions: The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification.[1] The SUT T1 CAS preemption signal generation is out of tolerance.[2] The SUT recognizes E1 and T1 CAS wink start signals greater than the maximum interval as valid.[3] During a remote busy condition on a T1 CAS or E1 CAS, the SUT takes approximately 5 minutes to change the status of the timeslots from an "In-Service/Active" state to a "Far-End-Busy" state.[4]
E1 CAS (DTMF, MFR1, DP)	Yes (Europe only)	Certified	Met all CRs and FRs with the following minor exceptions: The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification.[1] The SUT recognizes E1 and T1 CAS wink start signals greater than the maximum interval as valid.[3] During a remote busy condition on a T1 CAS or E1 CAS, the SUT takes approximately 5 minutes to change the status of the timeslots from an "In-Service/Active" state to a "Far-End-Busy" state.[4]
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all CRs and FRs with the following minor exceptions: The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification.[1] Failure to maintain busy out condition after restart messages are received from the distant switch.[5]
E1 ISDN PRI	No	Not Tested	The SUT offers an E1 ISDN PRI interface; however, this interface was not tested and is not covered under this certification. Since this is not a required interface for a SMEO or DVX, there is no operational impact.[6]
T1 SS7 (ANSI T1.619a)	No	Not Tested	This interface is not supported. Since this is not a required interface for a SMEO or DVX, there is no operational impact.[7]
E1 SS7 (ANSI T1.619a)	No	Not Tested	This interface is not supported. Since this is not a required interface for a SMEO or DVX, there is no operational impact.[7]
DSN Line Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs with the following minor exceptions: The precedence above ROUTINE ring cadence is not in accordance with GSCR specification.[8] The call pick-up feature does not pick-up the call with the highest precedence or longest ringing call first.[9] Three-way conference members do not maintain their assigned precedence levels.[10]
ISDN BRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all CRs and FRs with the following minor exceptions: The precedence above ROUTINE ring cadence is not in accordance with GSCR specification.[8] The call pick-up feature does not pick-up the call with the highest precedence or longest ringing call first.[9] Three-way conference members do not maintain their assigned precedence levels.[10]

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Table 1. SUT Interoperability Test Summary (continued)

DSN Line Interfaces (continued)			
Interface & Signaling	**Critical**	**Status**	**Remarks**
2-Wire Proprietary Digital	No	Certified	Met all CRs and FRs with the following minor exceptions: The precedence above ROUTINE ring cadence is not in accordance with GSCR specification.[8] The call pick-up feature does not pick-up the call with the highest precedence or longest ringing call first.[9] Three-way conference members do not maintain their assigned precedence levels.[10]
VoIP (IEEE 802.3u)	No	Certified	Met all CRs and FRs with the following minor exceptions: The precedence above ROUTINE ring cadence is not in accordance with GSCR specification.[8] The call pick-up feature does not pick-up the call with the highest precedence or longest ringing call first.[9] Three-way conference members do not maintain their assigned precedence levels.[10]
Voicemail			
Interface	**Critical**	**Status**	**Remarks**
2-Wire Proprietary Digital	No	Certified	Met all CRs and FRs.
Automated Call Distributor			
Interface	**Critical**	**Status**	**Remarks**
Internal	No	Certified	Met all CRs and FRs.
DSN Features and Capabilities			
Features and Capabilities	**Critical**	**Status**	**Remarks**
Common Features	No	Certified	Met all CRs and FRs with the following minor exception: Selective Call Rejection is not supported by the SUT.[11]
Attendant	No	Certified	Met all CRs and FRs with the following minor exception: The SUT attendant console does not support the automatic recall feature.[12]
Public Safety	Yes	Certified	Met all CRs and FRs with the following minor exception: Tandem call trace of a distant office DN is not supported by SUT.[13]
Preset Conferencing	No	Certified	This feature is met through the use of the Compunetx Contex(R) 240.
Nailed-up Connections	No	Not Tested	This feature is not supported. Since this is not a required feature for a SMEO or DVX, there is no operational impact.[7]
Precedence Access Threshold	No	Not Tested	This feature is not supported. Since this is not a required feature for a SMEO or DVX, there is no operational impact.[7]
DSN Hotline Services	Yes	Certified	The SUT met all CRs and FRs. Hotline Services is required only for analog interfaces. The SUT supports Hotline Services only with analog stations.
Network Management	Yes	Certified	Met all CRs and FRs with an IEEE 802.3u interface.
ISDN Services (EKTS)	Yes	Certified	Met all CRs and FRs with the following minor exceptions: When an EKTS member is assigned to an MLHG, a call to that EKTS member fails to ring the other EKTS members.[14] When an intercom call is placed on an EKTS station, the primary DN of the calling EKTS user is used and the station is made busy.[15]
Synchronization	Yes	Certified	Met all CRs and FRs.
Reliability	Yes	Certified	Met all CRs and FRs.
Security	Yes	See note 16.	See note 16.
VoIP			
Features and Capabilities	**Critical**	**Status**	**Remarks**
VoIP System	Yes	Certified	Met all CRs and FRs. The SUT is certified for VoIP with any VALAN or ASVALAN on the UC APL. See note 17.
Softphone	No	Certified	The SUT is certified with the Avaya IP Softphone Release 6.0 based upon Desktop Review 11.

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Table 1. SUT Interoperability Test Summary (continued)

Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, DP)	Yes	Certified	Met all CRs and FRs.
	T1 CAS (MFR1)	No	Certified	Met all CRs and FRs.
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Certified	Met all CRs and FRs.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all CRs and FRs.
	E1 ISDN PRI	No	Not Tested	The SUT offers an E1 ISDN PRI interface; however, this interface was not tested and is not covered under this certification. Since this is not a required interface for a SMEO or DVX, there is no operational impact.[6]
	Ground Start Line	Yes	Certified	Met all CRs and FRs.
DRSN	TPC 2-Wire analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs. See note 18.

NOTES:

1 The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification. The requirement states that the yellow alarm should be removed 15 seconds +/- 5 seconds upon DS1 restoration. The SUT removes the yellow alarm 30 seconds after the DS1 is restored. The operational impact is minor.

2 The SUT T1 CAS preemption signal generation is out of tolerance. The preemption signal generated by the SUT was measured 2 ms outside the GSCR required preemption signal of 345 ms +/- 5 ms. The operational impact is minor.

3 The SUT recognizes E1 and T1 CAS wink start signals greater than the maximum interval as valid. The SUT recognizes wink start signals from 100 ms to 395 ms as valid. The GSCR requirement specifies the wink start recognition range to be between 100 ms and 350 ms. The operational impact is minor.

4 During a remote busy condition on a T1 CAS or E1 CAS, the SUT takes approximately 5 minutes to change the status of the timeslots from an "In-Service/Active" state to a "Far-End-Busy" state. During this period of time, a ROUTINE call attempted over this span receives T-120 and precedence above ROUTINE call receives Blocked Precedence Announcement. After the state is changed, the correct treatment, an Isolated Code Announcement, is provided to all calls attempted over this span. The operational impact is minor.

5 When the SUT initiates a busy-out condition for a T1 PRI, and if the distant switch sends RESTART messages while the SUT has a busy-out condition, the SUT responds with RESTART ACKNOWLEDGEMENT messages; however, the SUT does not retransmit the SERVICE (Out-Of-Service) message for all of the busied channels. The result is that the distant switch idles the channels that the SERVICE (Out-Of-Service) messages were not retransmitted on. This condition can be eliminated by busying both ends. The operational impact is minor.

6 The SUT offers an E1 ISDN PRI interface; however, this interface was not tested and is not covered under this certification. Therefore, this interface is not authorized nor approved for use within the DSN. Since this is not a required interface for a SMEO or DVX, there is no operational impact.

7 The SUT does not support this. Since this is not required for a SMEO or DVX, there is no operational impact.

8 The precedence above ROUTINE ring cadence is not in accordance with GSCR specification. Since the cadence is different than a ROUTINE ring cadence, the operational impact is minor.

9 The SUT call pickup feature doesn't retrieve the call with the highest precedence first. The SUT retrieves unanswered call pickup group calls above ROUTINE in a random sequence. The GSCR requires that "If a call pickup group has more than one party in an unanswered condition and the unanswered parties are at different precedence levels, a call pickup attempt in that group shall retrieve the highest precedence call first." All unanswered precedence calls above ROUTINE in the pickup group do divert after 15-45 seconds if unanswered and are positively connected to either the attendant, night service, or alternate DN. The operational impact is minor.

10 Three-way conference members do not maintain their assigned precedence levels. Since the SUT class marks the conference members at the highest precedence level, the operational impact is minor.

11 Selective Call Rejection is not supported by the SUT. Since it is not a critical requirement for a SMEO or DVX, there is no operational impact.

12 The SUT attendant console does not support the automatic recall feature. The SUT does permit the attendant console to extend (camp-on) a caller to a busy station. Since the SUT provides this for the subscriber as a feature access code, the operational impact is minor.

13 Tandem call trace of a distant office DN is not supported by SUT. The operational impact is minor.

14 When an EKTS member is assigned to an MLHG, a call to that EKTS member fails to ring the other EKTS members. When a call is sent to an MLHG pilot number that causes an EKTS member to ring, all members of the EKTS group should have an incoming call appearance. The EKTS feature is certified as standalone and not when assigned as a member of an MLHG. MLHG interaction with EKTS is a conditional requirement; therefore, the operational impact is minor.

15 When an intercom call is placed on an EKTS station, the primary DN of the calling EKTS user is used and the station is made busy. In accordance with the GSCR specification, the EKTS intercom feature should not affect the busy/idle status of any of the DNs of the calling EKTS user. An EKTS station can have additional call appearances added to compensate for this discrepancy. The operational impact is minor.

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Table 1. SUT Interoperability Test Summary (continued)

NOTES (continued):

16   Security is tested by DISA-led Information Assurance test teams and published in a separate report.

17   An IPv6 capable system or product, as defined in the GSCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor Letter of Compliance signed by the Vice President of their company. The vendor stated, in writing, compliance to the following criteria by 31 December 2008:
(a) Conformant with IPv6 standards profile contained in the DISR.
(b) Maintaining interoperability in heterogeneous environments and with IPv4.
(c) Commitment to upgrade as the IPv6 standard evolves.
(d) Availability of contractor/vendor IPv6 technical support.

18   Interoperability Certification of the SUT does not constitute DRSN PM's approval for connectivity to the DRSN. It is the user's responsibility to request connectivity approval directly from the PM.

LEGEND:

802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	IPv4	Internet Protocol version 4
		IPv6	Internet Protocol version 6
ANSI	American National Standards Institute	ISDN	Integrated Services Digital Network
APL	Approved Products List	IT	Information Technology
ASVALAN	Assured Services Voice Application Local Area Network	LSSGR	Local Access and Transport Area (LATA) Switching System Generic Requirements
BRI	Basic Rate Interface	Mbps	Megabits per second
CAS	Channel Associated Signaling	MFR1	Multi-Frequency Recommendation 1
CRs	Capability Requirements	MLHG	Multi-Line Hunt Group
DISA	Defense Information Systems Agency	MLPP	Multi-Level Precedence and Preemption
DISR	DoD IT Standards Registry	ms	milliseconds
DN	Directory Number	NI 1/2	National ISDN Standard 1 or 2
DoD	Department of Defense	PM	Program Manager
DP	Dial Pulse	PRI	Primary Rate Interface
DRSN	Defense Red Switch Network	PSTN	Public Switched Telephone Network
DSN	Defense Switched Network	SMEO	Small End Office
DS1	Digital Signal Level 1	SS7	Signaling System 7
DSS1	Digital Subscriber Signaling 1	SUT	System Under Test
DTMF	Dual Tone Multi-Frequency	T1	Digital Transmission Link Level 1 (1.544 Mbps)
DVX	Deployable Voice Exchange	T1.607	ISDN – Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
E1	European Basic Multiplex Rate (2.048 Mbps)		
EKTS	Electronic Key Telephone System	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
FRs	Feature Requirements	TPC	Twisted Pair Copper
GR	Generic Requirement	UC	Unified Capabilities
GR-506-CORE	LSSGR: Signaling for Analog Interfaces	VALAN	Voice Application Local Area Network
GSCR	Generic Switching Center Requirements	VoIP	Voice over Internet Protocol
IEEE	Institute of Electrical and Electronics Engineers		

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Table 2. SMEO Requirements

DSN Trunk Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 SS7 (ANSI T1.619a) E1 SS7 (ITU-T Q.735.3) T1 CAS (MFR1) T1 CAS (DTMF, DP) E1 CAS (MFR1, DTMF, DP) T1 ISDN PRI NI 1/2 (ANSI T1.619a) E1 ISDN PRI (ITU-T Q.955.3)	No No (Europe only) No Yes Yes (Europe only) Yes No (Europe only)	Trunking	• Framing (R) • Line Code (R) • Signaling (R) • Alarms (R) • WWNDP (R) • Out pulsing digit formats (R: CAS only) • Routing (R) • Trunk Groups (R) • Call Processing (R) • CAS to CCS trunk interworking (C) • PCM-24/PCM-30 Interoperation (R) • Direct Inward Dialing (C)	• GSCR Sect. 7 • GSCR Sect. 7 • GSCR Sect. 5 • GSCR Sect. 2.5.7, 7.1.4 & 7.2.2 • GSCR Sect. 4.5.1 • GSCR Sect. 4.5.2 • GSCR Sect. 4.2 • GSCR Sect. 2.5.5 & 2.5.6 • GSCR Sect. 4 • GSCR Sect. 3.10 • GSCR Sect. 7.3 • GSCR Sect. 2.3.2
		Voice	• MOS (R) • MLPP (R) • Secure calls (R)	• CJCSI 6215.01B • GSCR Sect. 3 • CJCSI 6215.01B
		Facsimile	• Analog: TIA/EIA-465-A (R)	• DISR
		Data	• Modem (VBD) (R) • 56 kbps switched data (R: PRI only) • 64 kbps switched data (R: PRI only) • NX56 synchronous BER (R: PRI only) • NX64 synchronous BER (R: PRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01B • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • CJCSI 6215.01B
		VTC	• ITU-T H.320 (R: PRI only)	• DISR
DSN Line Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
2W Analog ISDN BRI NI 1/2 (ANSI T1.619a) 2W Digital Proprietary VoIP (IEEE 802.3u)	Yes Yes No No	Access	• DN Identification (R) • Line signaling (R) • Loop Start Line (R: 2-Wire Analog only) • Ground Start Line (R) • Alerting Signals and Tones (R) • WWNDP (R) • Call Processing (R) • Call Treatments (R) • 2W user access (R: 2-Wire Analog only) • Analog busy/idle (R: 2-Wire Analog only)	• GSCR Sect. 2.1.1 • GSCR Sect. 5.2 • GSCR Sect. 5.2.1 • GSCR Sect. 5.2.2 • GSCR Sect. 5.5 • GSCR Sect. 4.5 • GSCR Sect. 4.4 • GSCR Sect. 4.1 • GSCR Sect. 4.3.3 • GSCR Sect. 4.3.4.1
		Voice	• MOS (R) • Announcements (R) • MLPP (R) • Secure Calls (R)	• CJCSI 6215.01B • GSCR Sect. 3.1.3 • GSCR Sect. 3.4.3/3.9 • CJCSI 6215.01B
		Facsimile	• Analog: TIA/EIA-465-A (R)	• DISR
		Data	• Modem (VBD) (R) • 56 kbps switched data (R) • 64 kbps switched data (R: BRI only) • NX56 synchronous BER (R: BRI only) • NX64 synchronous BER (R: BRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01B • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • CJCSI 6215.01B
		VTC	• ITU-T H.320 (R: BRI only)	• DISR

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Table 2. SMEO Requirements (continued)

SUT Voice Mail Interfaces			
Interface	**Critical**	**Requirements Required or Conditional**	**References**
2W Digital Proprietary	No	• FCC Part15/Part 68 (R): Analog only • DTMF out pulsing (C) • DISR compliance as applicable (R) • ROUTINE precedence only in accordance with GSCR, Section 3.3 (R) • TIA/EIA-470-B (R): Analog only	• GSCR A7.5 • GSCR A7.5, 5.4.1, 5.4.2 • GSCR A7.5 • GSCR A7.5.5 • GSCR A7.5.1
Automated Call Distributor Interfaces			
Interface	**Critical**	**Requirements Required or Conditional**	**References**
Internal	No	• DTMF out pulsing (C) • DISR compliance as applicable (R) • ROUTINE precedence only in accordance with GSCR, Section 3.3 (R)	• GSCR Sect. A7.5, 5.4.1, 5.4.2 • GSCR Sect. A7.5 • GSCR Sect. A7.5
DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Common Features	Yes	• Selective call rejection (C) • Denied originating service (C) • Code restriction and diversion (R) • Call waiting (C) • Three-way calling (C) • Add-on transfer and conference calling and call hold (C) • Call forwarding (C) • Call pick-up (C)	• GSCR Sect. 2.1.2 • GSCR Sect. 2.1.3 • GSCR Sect. 2.1.4 • GSCR Sect. 2.1.5 • GSCR Sect. 2.1.6 • GSCR Sect. 2.1.7 • GSCR Sect. 2.1.8 • GSCR Sect. 2.1.9
Attendant	No	• Initiate all precedence levels (C) • Visual display (C) • Override class of service (C) • Override busy line (C) • Call deflection (C) • Auto recall (C) • Waiting queue (C)	• GSCR Sect. 2.2.1 • GSCR Sect. 2.2.2 • GSCR Sect. 2.2.3 • GSCR Sect. 2.2.4 • GSCR Sect. 2.2.5 • GSCR Sect. 2.2.6 • GSCR Sect. 2.2.7
Public Safety	Yes	• Basic Emergency Service (911) (C) • Trace of terminating calls (R) • Outgoing call trace (R) • Tandem call trace (R) • Trace of a call in progress (R)	• GSCR Sect. 2.4.1 • GSCR Sect. 2.4.2 • GSCR Sect. 2.4.3 • GSCR Sect. 2.4.4 • GSCR Sect. 2.4.5
Preset Conferencing	No	• Support 10 bridges; 1 originator and 20 conferees per bridge (C) • Assign up to 20 address numbers per bridge (C) • Use KXX codes for bridge access (C) • Conference notification recorded announcement (C) • Auto retrial and alternate address (C) • Bridge release (C) • Lost connection (C) • Secondary conferencing (C) • Address translation (C)	• GSCR Sect. 2.6 • GSCR Sect. 2.6 • GSCR Sect. 2.6 • GSCR Sect. 2.6.1 • GSCR Sect. 2.6.2 • GSCR Sect. 2.6.3 • GSCR Sect. 2.6.4 • GSCR Sect. 2.6.5 • GSCR Sect. 2.7
Nailed-up Connections	No	• Between any two like terminations (C) • PCM-24 and PCM-30, both CAS and CCS (C) • Supervision passed end-to-end for A/D or D/A (C) • Monitored and auto reconfigure (C) • Support at least 10% of circuits as nailed-up (C) • Non-preemptable (C)	• GSCR Sect. 2.8 • GSCR Sect. 2.8 • GSCR Sect. 2.8 • GSCR Sect. 2.8 • GSCR Sect. 2.8 • GSCR Sect. 2.8

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Table 2. SMEO Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	Critical	Requirements Required or Conditional	References
PAT	No	• Class mark for/not for PAT screening (C) • 7 PAT mechanisms (C) • Outgoing call screening (C) • Functional structure (C) • Simultaneous calls limitation (C) • Overflow process (C) • Decrementing call-in-progress count (C) • Call treatment (C) • Queuing (C) • Attendant calls (C) • Operations measurement registers (C) • Maintenance and Administration of thresholds (C)	• GSCR Sect. 2.11.1 • GSCR Sect. 2.11.1 • GSCR Sect. 2.11.1.1 • GSCR Sect. 2.11.1.2 • GSCR Sect. 2.11.1.3 • GSCR Sect. 2.11.1.4 • GSCR Sect. 2.11.1.5 • GSCR Sect. 2.11.1.6 • GSCR Sect. 2.11.1.7 • GSCR Sect. 2.11.1.8 • GSCR Sect. 2.11.1.9 • GSCR Sect. 2.11.1.10
DSN Hotline Services	Yes	• Hotline restrictions (R) • Auto initiate (R) • Analog and digital (R) • Subscription basis (R) • Protected hotline calling (R) • WWNDP interoperable (R)	• GSCR Sect. 2.12 • GSCR Sect. 2.12 • GSCR Sect. 2.12 • GSCR Sect. 2.12 • GSCR Sect. 2.12.1-4 • GSCR Sect. 2.12.5
Network Management	Yes	• Interfaces (R) • Measurements and data generation (R) • Fault management (R) • Configuration management (R) • Accounting management (R) • Performance management (R) • Network Management controls (R) • Remote access (R)	• GSCR Sect. 9.1 • GSCR Sect. 9.2 • GSCR Sect. 9.3 • GSCR Sect. 9.4 • GSCR Sect. 9.5 • GSCR Sect. 9.6 • GSCR Sect. 9.7 • GSCR Sect. 9.8
ISDN Services	No	• EKTS (C)	• GSCR Sect. 10, Table 10-3
Synchronization	Yes	• Line timing mode (R) • Internal Stratum 4 (R)	• GSCR Sect. 11.1.1.2 • GSCR Sect. 11.1.2.2
Reliability	Yes	• GR-512-CORE (R)	• GSCR Sect. 12
Security	Yes	• GR-815, STIGs, and DIACAP (replacement for DITSCAP) (R)	• GSCR Sect. 13

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Table 2. SMEO Requirements (continued)

VoIP			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
VoIP System	No	VoIP function is conditional. If VoIP is provided, **all** of the following requirements must be met: • Voice Quality with MOS of 4.0 or better • Class of Service (CoS) and Quality of Service (QoS) • ITU-T G.711 PCM Codec • Traffic Engineering • Security • Network Management • Line timing • Internal Clock • Latency ≤ 60 milliseconds • Packet Loss • IPv6 capable	 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR paragraph 1.7
IP Softphone	No	• Voice Features and Capabilities IAW Section 5.3.2.2.2.1 • System Availability IAW Section 5.3.2.5.2.1 • Voice Instrument IAW Section 5.3.2.6.1 • Tones and Announcements IAW Section 5.3.2.6.1.1 • Audio Codecs IAW Section 5.3.2.6.1.2 • Handset Requirements IAW Section 5.3.2.6.1.3 • VoIP Sampling Standard IAW Section 5.3.2.6.1.4 • Authentication to LSC IAW Section 5.3.2.6.1.5 • End Instrument to ASLAN Interface IAW Section 5.3.2.6.3 • Network Infrastructure End-toEnd Performance Requirements IAW Section 5.3.3 • VVoIP DSCP IAW Section 5.3.3.3.2 • Information Assurance Requirements IAW Section 5.4	• UCR 2008 Change 1, para. 5.3.2.6.1.7 • UCR 2008 Change 1, para. 5.3.2.6.1.7 • UCR 2008 Change 1, para. 5.3.2.6.1.7 • UCR 2008 Change 1, para. 5.3.2.6.1.7 • UCR 2008 Change 1, para. 5.3.2.6.1.7 • UCR 2008 Change 1, para. 5.3.2.6.1.7 • UCR 2008 Change 1, para. 5.3.2.6.1.7 • UCR 2008 Change 1, para. 5.3.2.6.1.7 • UCR 2008 Change 1, para. 5.3.2.6.1.7 • UCR 2008 Change 1, para. 5.3.2.6.1.7 • UCR 2008 Change 1, para. 5.3.2.6.1.7 • UCR 2008 Change 1, para. 5.3.2.6.1.7
Network Gateways			
Gateway	**Critical**	**Requirements Required or Conditional**		**References**
PSTN[1]	Yes	Trunking	• Positive Identification Control (R) • On-Netting (R) • Off-Netting (R)	• CJCSI 6215.01B • CJCSI 6215.01B • CJCSI 6215.01B
DRSN[2]	Yes	Access	• Alerting Signals and Tones (R) • Call Processing (R) • Call Treatments (R) • Analog busy/idle (R)	• GSCR Sect. 5.5 • GSCR Sect. 4.4 • GSCR Sect. 4.1 • GSCR Sect. 4.3.4.1
		Voice	• MOS (C) • MLPP (C) • Secure calls (C)	• CJCSI 6215.01B • GSCR Sect. 3 • CJCSI 6215.01B

NOTES:
1   Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.
2   Facsimile, data, and VTC services are not provided via the DSN to DRSN interface.

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Table 2. SMEO Requirements (continued)

LEGEND:

2W	2-Wire	GR	Generic Requirement (Telcordia)	PAT	Precedence Access Threshold
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	GR-512-CORE	LSSGR: Reliability, Section 12	PCM	Pulse Code Modulation
A	Appendix	GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security	PCM-24	Pulse Code Modulation - 24 Channels
A/D	Analog to Digital Conversion			PCM-30	Pulse Code Modulation - 30 Channels
ANSI	American National Standards Institute	GSCR	Generic Switching Center Requirements	PRI	Primary Rate Interface
App.	Appendix	H.320	Standard for Narrowband VTC	PSTN	Public Switched Telephone Network
ASLAN	Assured Services Local Area Network	IAW	in accordance with	Q.735.3	SS7 Signaling Standard for E1 MLPP
BER	Bit Error Ratio	IEEE	Institute of Electrical and Electronics Engineers, Inc.	Q.955.3	ISDN Signaling Standard for E1 MLPP
BRI	Basic Rate Interface				
C	Conditional	IP	Internet Protocol	QoS	Quality of Service
CAS	Channel Associated Signaling	IPv6	Internet Protocol version 6	R	Required
CCS	Common Channel Signaling	ISDN	Integrated Services Digital Network	Sect.	Section
CJCSI	Chairman of the Joint Chiefs of Staff Instruction	IT	Information Technology	SMEO	Small End Office
		IT	Information Technology	SS7	Signaling System 7
CoS	Class of Service	ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	STE	Secure Terminal Equipment
D/A	Digital to Analog Conversion			STIGs	Security Technical Implementation Guides
DIACAP	DoD Information Assurance Certification and Accreditation Process			STU-III	Secure Telephone Unit – 3rd Generation
DISR	DoD IT Standards Registry	LSC	Local Session Controller		
DITSCAP	DoD IT Security Certification and Accreditation Process	LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements	T1	Digital Transmission Link Level 1 (1.544 Mbps)
DN	Directory Number			T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
DoD	Department of Defense				
DP	Dial Pulse	kbps	kilobits per second	TIA	Telecommunications Industry Association
DRSN	Defense Red Switch Network	KXX	K= any number 2-8; X= any number 1-9		
DSCP	Differentiated Services Code Point			TIA/EIA-465-A	Group 3 Facsimile Apparatus for Document Transmission
		Mbps	Megabits per second	TIA/EIA-470-B	Performance and Compatibility Requirements for Telephone Sets with Loop Signaling
DSN	Defense Switched Network	MFR1	Multi-Frequency Recommendation 1		
DTMF	Dual Tone Multi-Frequency				
E1	European Basic Multiplex Rate (2.048 Mbps)	MLPP	Multi-Level Precedence and Preemption		
EKTS	Electronic Key Telephone System	MOS	Mean Opinion Score	VBD	Variable bit data
		NI 1/2	National ISDN Standard 1 or 2	VoIP	Voice over Internet Protocol
EIA	Electronic Industries Alliance			VTC	Video Teleconferencing
FCC	Federal Communications Commission	NX56	Data format restricted to multiples of 56 kbps	VVoIP	Voice and Video over IP
G.711	Standard for PCM of Voice Frequencies	NX64	Data format restricted to multiples of 64 kbps	WWNDP	Worldwide Numbering and Dialing Plan

Table 3. SUT SMEO/DVX Requirement Differences and Interoperability Status

GSCR Paragraph	Requirement	SMEO Critical	DVX Critical	Status	Remarks
2.3.3	NI 1/2 BRI	No	Yes	Certified	Met all critical CRs and FRs.
A2.5.2.1	Preset Conferencing	No	Yes	Certified	Met all critical CRs and FRs.
2.11.1.10	Maintenance and Administration of Thresholds	No	Yes	Certified	Met all critical CRs and FRs.
2.12	DSN Hotline Service	Yes	No	Certified	Met all critical CRs and FRs.
3.6	ISDN BRI MLPP interactions	Yes	No	Certified	Met all critical CRs and FRs.
4.3.1	E&M Lead Signaling States	No	Yes	Certified	Met all critical CRs and FRs.
4.3.2	Four Wire E&M Analog User Access Lines	No	Yes	Certified	Met all critical CRs and FRs.
4.5.1.8	Emergency Service 911 Conflict Resolution	Yes	No	Certified	Met all critical CRs and FRs.

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Table 3. SUT SMEO/DVX Requirement Differences and Interoperability Status (continued)

GSCR Paragraph	Requirement	SMEO Critical	DVX Critical	Status	Remarks
Table 4-9	DSN Switch MFR1 Out pulsing Digit Format	No	Yes	Certified	Met all critical CRs and FRs.
Table 4-10	DSN Switch DTMF Out pulsing Digit Format	No	Yes	Certified	Met all critical CRs and FRs.
5.1	Network Power Systems for External Interfaces	Yes	No	Certified	Met all critical CRs and FRs.
5.4.3	MFR1 2/6 Signaling	No	Yes	Certified	Met all critical CRs and FRs.
5.7.1.2.1	S/T Reference Point	Yes	No	Certified	Met all critical CRs and FRs.
5.7.1.3	Data-Link Layer	Yes	No	Certified	Met all critical CRs and FRs.
5.7.1.3.1	Data-Link Connections	Yes	No	Certified	Met all critical CRs and FRs.
5.7.1.3.2	Peer-to-Peer Procedures of the Data-Link Layer	Yes	No	Certified	Met all critical CRs and FRs.
5.7.1.4	Layer 3 DSN User-to-Network Signaling	Yes	No	Certified	Met all critical CRs and FRs.
5.7.1.4.2	DSN User-to Network Signaling for CS Bearer Service	Yes	No	Certified	Met all critical CRs and FRs.
5.7.1.4.3	Sequence of Messages for DSN CS Calls	Yes	No	Certified	Met all critical CRs and FRs.
5.7.1.4.4	Message Functional Definitions and Content	Yes	No	Certified	Met all critical CRs and FRs.
5.7.1.4.5	General Message Format and Information Elements Coding	Yes	No	Certified	Met all critical CRs and FRs.
9.5.1	DSN Settable CDR Fields	Yes	No	Certified	Met all critical CRs and FRs.
Section 12	Reliability	Yes	No	Certified	Met all critical CRs and FRs.
Section 13	Security	Yes	No	Certified	Met all critical CRs and FRs.

NOTE: The requirements for SMEOs and DVXs are identical except for those listed in above.

LEGEND:
A	Appendix	FRs	Feature Requirements
BRI	Basic Rate Interface	GSCR	Generic Switching Center Requirements
CDR	Call Detail Recording	ISDN	Integrated Services Digital Network
CRs	Capability Requirements	MFR1	Multi-Frequency Recommendation 1
CS	Circuit Switched	MLPP	Multi-Level Precedence and Preemption
DSN	Defense Switched Network	NI 1/2	National ISDN Standard 1 or 2
DTMF	Dual Tone Multi-Frequency	SMEO	Small End Office
DVX	Deployable Voice Exchange	S/T	Four-wire ISDN BRI interface
E&M	Ear and Mouth	SUT	System Under Test

5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet), or http://199.208.204.125 (SIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly through government civilian or uniformed military personnel from the Unified Capabilities Certification Office (UCCO), e-mail: ucco@disa.mil.

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FOR THE COMMANDER:

Enclosure a/s   for RICHARD A. MEADOR
                                               Chief
                                               Battlespace Communications Portfolio

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U.S. Joint Forces Command, Net-Centric Integration, Communication, and Capabilities Division, J68
Defense Information Systems Agency, GS23

ADDITIONAL REFERENCES

(c)   Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01B, "Policy for Department of Defense Voice Services," 23 September 2001

(d)   Joint Interoperability Test Command, Memo, JTE, "Special Interoperability Test Certification of the Avaya S8710 and S8720 Digital Switching Systems with Software Release Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)," 2 October 2007

(e)   Defense Information Systems Agency, "Department of Defense Voice Networks Generic Switching Center Requirements (GSCR), Errata Change 2," 14 December 2006, Revised 27 March 2007

(f)   Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006

(g)   Office of the Assistant Secretary of Defense, "Department of Defense Unified Capabilities Requirements 2008 Change 1," 22 January 2010

Enclosure